REMARKS

In view of both the amendments presented above and the following discussion, the Applicant submits that none of the claims now pending in the application is obvious under the provisions of 35 USC § 103. Thus, the Applicant believes that all of these claims are now in allowable form.

If, however, the Examiner believes that there are any unresolved issues requiring adverse final action in any of the claims now pending in the application, the Examiner should telephone Mr. Peter L. Michaelson, Esq. at (732) 542-7800 so that appropriate arrangements can be made for resolving such issues as expeditiously as possible.

Specification amendment

The specification has now been amended at page 6, lines 9 and 10 simply to define a term "returned artifact signal" which now appears in the claims in order to collectively and easily refer to a return signal that may contain any of the specific signals e1-e5 listed in page 6, lines 1-8. Such a definition does not constitute new matter.

Status of claims

Claims 21, 26, 33, 34, 36 and 37 have each been amended.

Rather than re-writing claim 32 to amend various recitations and also to simplify the Examiner's consideration of the ensuing claim, independent claim 32 has simply been replaced by independent claim 41 which incorporates these amendments. Claims 33, 34 and 36 have now all been amended to depend from claim 41.

Claims 42-55 are new.

Rejection under 35 USC § 103

The Examiner rejected claims 21-40, as they stood prior to this amendment, under the provisions of 35 USC § 103 as being obvious in view of the teachings in the Karlsen et al application (International patent application publication WO 97/15124) taken in view of the teachings in the Hollier application (International patent application publication WO 94/00922). With respect to the present claims, this rejection is respectfully traversed. To simplify the ensuing discussion, this rejection will now be discussed primarily with respect to independent claim 21, as it now stands.

In particular, the Examiner states that the Karlsen et al application discloses various elements of claim 21, as it previously stood, including combining a talker speech signal and a returned signal to yield a combined speech signal and then subjecting the combined speech signal to a quality measurement. The Examiner concedes that the Karlsen et al application does not specifically teach the implementation of an objective

measurement of perceptual quality of speech signals which results in an output signal that represents an estimated value for talking quality. However, she states that implementation of objective speech quality measurements of telecommunications equipment was well known in the art in order to detect degradation of signals transmitted over communication links. In that regard, see points to the teachings of the Hollier application as describing an approach for making such measurements. Given this, she opines that it would have been obvious, at the time the present invention was made, to modify the system taught by the Karlsen et al application to include the technique taught by the Hollier application in order to detect any degradation of the transmitted speech and verify the quality of the adaptation of the filter (adaptive filter 12) taught by the Karlsen et al application and improve the operation of that filter -- and any one of skill at the time in the art would recognize the advantages in having done so. Thus, the Examiner apparently concludes that, as a result of making such modifications, the Applicant's claimed invention would result. As to the invention recited in present claim 21 (as well as in all the other independent claims), the Examiner's view is incorrect.

As discussed in the Applicant's prior amendment mailed January 11, 2006, the Karlsen et al application is directed to a telephone-link circuit, specifically at a listener's side, which includes an echo-minimizing device, i.e., echo canceller, for reducing electrical echo caused in a four-to-two wire conversion in a PSTN/subscriber interface.

Consider first, the prior art arrangement shown in FIG. 2, upon which the embodiments in FIGs. 3 and 4, are based. FIG. 2 is a functional representation of both near-end hybrid 10 and echo cancellation of far-end generated echo. Referring to FIG. 1 -- which is a rather simplistic view of a conventional bi-directional telecommunications channel, this echo is of far-end generated echo of near-end speech (from subscriber B) and near-end noise that has leaked through the far-end hybrid and then collectively carried back over the communication channel (top path in FIG. 1) to the near-end hybrid and there through to subscriber A. For simplicity, all such noise will be ignored hereinafter.

As expressly depicted in FIG. 2 and discussed in page 4, line 7 et seq, far-end speech x(n) will generate echo at the near-end as a result of the near-end hybrid (in a functionally identical fashion through which near-end speech will generate echo at the far-end due to the far-end hybrid), that process being represented by transfer function 10, i.e. $(H(q^{-1}))$. The resulting near-end echo s(n) is combined with near-end speech v(n) through adder 14 to yield output signal y(n). If subscriber B is listening at the time, then signal y(n) will primarily contain near-end generated echo, i.e. that generated by hybrid 10.

Now, with this in mind, a near-end echo canceller, formed of adaptive filter 12 having a transfer function of $H^{\wedge}(q^{-1})$, is used to generate an echo replica (estimate) of the near-end generated echo. This estimate, $s^{\wedge}(n)$, is then subtracted, via summer 16, from output signal y(n) to yield an error signal e(n). This error signal, applied as a

feedback signal, controls the adaptation of filter 12, i.e. the change in its tap coefficients, such that the filter adapts its operation (varies its tap coefficients) in such a manner that, over time, the near-end generated echo replicas accurately mimic the actual near-end echo values, hence driving error signal e(n) to zero. This error signal is also then returned to the far-end for use thereat. It is important to note that the same near-end functionality shown in FIG. 2 exists at the far-end, per page 4, lines 9-10 of the specification which state: "(a similar arrangement is provided at the far end side)".

Curiously, the Karlsen et al specification does not address at all how this specific error signal e(n) is used at the far-end along with the corresponding error signal locally generated there by a far-end summer corresponding to near-end summer 16. Note that the far-end error generated error signal is not shown as an input to the near-end functional blocks shown in FIG. 2, only the near-end error signal is. But what is indisputably clear from FIG. 2 is that signal returned by the circuitry at each end of the channel to the other is an <u>ERROR SIGNAL</u> and nothing more.

Therefore, what the Examiner must keep squarely in mind from the clear, unambiguous and express teachings of the Karlsen et al application is the nature of the returned signal: it is an <u>ERROR</u> signal (e(n)); it is <u>not</u> a speech signal or even a speech signal that has been modified (such as by echo) by passage through the communication channel.

FIG. 3, as described in page 4, line 36 et seq, shows a known modification of the conventional arrangement depicted in FIG. 2. Here too, the same signal x(n) is shown, with the difference, between the arrangements shown in FIGs. 2 and 3, being that the latter arrangement also contains programmable filter 18 to yield a dual-filter echo canceller. This particular arrangement is designed to handle double-talk scenarios, i.e. where in contrast to the arrangement in FIG. 2 which is designed to handle speech originating from only one end of the channel at a time and perform cancellation at the listener's side, both the called and calling parties are simultaneously talking. When double-talk occurs, error signal e(n) will also contain near-end speech components, hence causing the adaptation process in filter 12 to diverge and thus producing erroneous tap coefficients. To ameliorate this problem, the conventional approach taken in FIG. 3 is to include programmable filter 18 which provides actual echo cancellation but which is not driven by the error signal that drives adaptive filter 12. During non-double talk intervals, the tap coefficients of filter 12 are transferred to filter 18 for use therein. During double-talk intervals, no coefficient transfer takes place, hence programmable filter 18 uses the coefficients that were transferred to it immediately before the onset of the double-talk interval. As can be clearly seen and as expressly described in page 5, line 5 et seq, the arrangement in FIG. 3, just like that in FIG. 2, also returns an error signal, here being $e_f(n)$. This error signal is simply the difference between the output signal y(n), which under non-double talk intervals is the actual near-end echo values,

and their near-end echo replicas (estimates) generated by filter 18.

FIG. 4 depicts the inventive modification over the conventional approach shown in FIG. 3. This modification was necessitated by the fact that some conditions exist, such as high background noise environments, where the tap coefficients produced by adaptive filter 12 may not be better than those then existing in programmable filter 18 but are nonetheless transferred to filter 18.

To the extent relevant and described in page 7, line 7 et seq of the Karlsen et al specification, the modification involves, in essence, implementing two near-end echo cancellers: one having adaptive filter 12 and the other having programmable filter 18. Both cancellers operate independently of each other and in parallel, with each generating its own near-end replica (estimate) signal ($s\hat{}_p(n)$ and $s\hat{}_a(n)$). Decision logic 24 decides, in any instance, which filter to use in generating a returned error signal. In particular, the echo replicas generated by each of the filters is applied to inputs of a different one of corresponding summers 16 and 22, which subtract those replicas from output signal y(n) to generate two separate residual error signals, $e_a(n)$ and $e_p(n)$, respectively. Through selection criteria, which is not particularly relevant to the present discussion, logic 24 selects, using quality measures and for each different sample n, which filter is then "better" than the other. These quality measures are based on correlation of energy content (E) of the residual error signals. Decision logic 24 generates

error signal e(n) as containing the residual error signal (difference between actual near-end echo values and corresponding near-end echo replicas) then based on the echo replica provided by the "best" filter.

Again, as with the other embodiments shown in FIGs. 2 and 3, the apparatus shown in FIG. 4 at the near-end returns an ERROR SIGNAL (e(n)) to its peer apparatus at the far-end.

Thus, as the Karlsen et al patent unequivocally teaches, the signal returned by the echo canceller at each end of a telecommunications channel to its peer canceller at the other end is an ERROR SIGNAL, nothing else.

The Hollier application, as the Examiner correctly notes, teaches an arrangement for use with telecommunications apparatus for providing an objective measure of speech quality, i.e. degradation to transmitted speech, that occurs as a result of transmission of that speech through the apparatus and as would be perceived by human listeners. It does so by modelling the perceptual properties of human hearing.

In essence, to the extent relevant and as also discussed in the Applicant's prior amendment mailed January 11, 2006, the Hollier application discloses a test apparatus (specifically apparatus 4 as shown in FIG. 1) which supplies a speech-like test signal to input 2 of telecommunications equipment 1 to be tested. This apparatus then receives a corresponding distorted test signal from

output 3 of the equipment, and determines a difference in perceptibility ('to a human listener') of the distorted test signal with respect to a corresponding original test signal. As shown in FIG. 2 and discussed in page 7, line 21 et seq, test apparatus 4 includes signal generator 7 and signal analyzer 8. The signal generator supplies the speech-like signal, while the signal analyzer, using a perceptual model of human hearing, analyzes the signal received from the apparatus under test. As noted in page 11, line 21 et seq, the analyzer produces an "acceptability" output signal which reflects the distortion of the test signal as perceived by a human listener (given the response of the human ear as then understood and modelled), hence, in effect, predicting *a* listening quality of a one-way speech signal through the telecommunications equipment under test. See page 13, lines 19-26.

As is clearly shown in FIG. 2, test apparatus 4 taught by the Hollier application generates one output on lead 5: a test signal from signal generator 7, and has one input on lead 6: the response from the equipment under test applied as an input to analyzer 8. Within apparatus 4, analyzer 8 also receives, as another input and via lead 9, the same test signal generated by signal generator 7. Analyzer 8, through use of its perceptual hearing model, effectively compares the test signal with the response to it provided by the apparatus in order to ascertain the differences there between as perceived by a human listener. Using these differences, analyzer 8 determines and provides a quality output signal on output port 10.

The Applicant's present invention is directed to measuring a so-called *talking quality* of a telephone link in a bi-directional channel (forward and return directions) in a telecommunications network, e.g., the influence of echo on the perceptual quality *on the talker's side* of that link. See, page 2, line 37 through page 3, line 2 of the present specification. This sharply contrasts with the approach taught by the Karlsen et al application which focuses on the listener's side.

To do so and as the present Applicant teaches, an original talker speech signal s(t) and a combined signal ((s'(t)) are both fed as inputs to an objective measurement device, such as PSQM (perceptual speech quality measurement) system 32 shown in FIG. 3 of the present application, for obtaining an output signal (q(t)) that represents an estimated value of the *perceptual talking quality*. The combined signal is a combination of a returned signal (r(t)) originating from the network and the original talker speech signal. The return signal results from transmission of the original talker speech signal in the forward channel through the network and, as a result, appears in a return channel but with various other signals (which may be viewed in some instances as impairments) imparted by the channel. These other signals include, as indicated on page 5, line 39 to page 6, line 8 of the present specification (with reference to FIG. 2 of the present application):

a) a signal e1 representing acoustic echo;

b) a signal e2 representing an electrical echo possibly in combination with the acoustical echo;

c) a signal e3 which represents the signal e2 as affected, i.e. delayed or distorted, by the network 20;

d) a signal e4 which represents the signal e3 in combination with a sidetone signal; and

e) a signal e5 which is an acoustic equivalent of the signal e4.

For ease of reference, the Applicant will use the term "returned artifact signal" to refer to the return signal that may contain any or all of these "other" signals which result from transmission of the original talker speech signal through the forward channel of a telecommunications link.

The resulting output signal (q(t)), which represents an estimate of the perceptual talking quality, may then be used to control an echo-minimizing device, as is typically included within an established telephone link. See, e.g., page 7, lines 37-39.

The returned artifact signal is clearly <u>not</u> an error signal, but rather a modified version of the original talker speech signal. In that regard, the returned artifact signal is <u>not</u> a difference between actual echo values and corresponding echo replicas -- which is the only return signal which the Karlsen et al application explicitly teaches, or any difference-based signal for that matter. Such an error signal would clearly <u>not</u> qualify or be useable as an input to a perceptual speech quality measurement (PSQM) system for the simple reason that, in instances where the actual echo values are quite large but echo cancellation is accurate, the error signal, being a difference, will tend

towards zero, thus causing significant error in the PSQM process were such that signal to be applied as an input thereto.

Advantageously and in accordance with the present invention, the Applicant has discovered that by applying both the original talker signal (appearing in the forward channel) and the combined signal, including both that talker signal and the returned artifact signal resulting thereby, to separate inputs of a PSQM system, that system can objectively and accurately determine a quality measurement of the returned artifact signal, as would be perceived by a human listener, relative to the original talker signal. It is simply axiomatic that a quality measurement can only be made on a relative basis. Here, the undistorted (unmodified) version of the original speech signal is applied to the PSQM system as a reference or standard against which the returned artifact signal, also applied to the PSQM system, is effectively compared. Without comparing two similar signals, such as those taught by the applicant, an accurate quality measurement simply can not be made.

Given this, does the combination of the teachings in the Karlsen et al and Hollier applications lead to the present invention? No.

The Examiner cites to the holding in *In re Fine* (837 F.2d 1071, 5 USPQ2sd 1596 (Fed. Cir. 1988) and *In re Jones* 958 F.2d 347, 21 USPQ2d 1941 (Fed Cir. 1992) by stating: "obviousness can only be established by combining or modifying the teachings of the prior art to produce the

claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art". With this in mind, the Examiner opines that such a person would think to combine the teachings of the Karlsen et al and Hollier applications in order to more accurately detect degradation of signals transmitted through a communications system and thus improve the performance of the adaptive filter taught by the Karlsen application. But even if one were to combine these teachings as the Examiner posits, would the result anticipated by the Examiner occur? No. Why not? Simply because, on further examination, it becomes patently apparent that an approach premised on those combined teachings would frustrate the very purpose the Examiner seeks to attain: improved performance.

Unfortunately, the Examiner has not provided any insight whatsoever as to how she would specifically combine those teachings to integrate the perceptual test system taught by the Hollier application into the echo cancellers taught by the Karlsen et al application. Nevertheless, the Applicant believes that those skilled in the art would envision only one possible approach to do so. Under close scrutiny and as the Examiner will soon see, that approach fails.

Turning to the drawings provided as Exhibits to this amendment, Exhibit A, as a starting point, simply depicts the conventional echo cancelling arrangement as shown in FIG. 2 of the Karlsen et al application. Exhibit A contains no modifications.

Exhibit B shows the conventional approach shown in FIG. 2 but modified to incorporate the Hollier test apparatus. In order to control the performance of the echo canceller, specifically adaptation of the tap coefficients of adaptive filter 12, and thus ostensibly improve its performance -- as the Examiner proposes, the Hollier test apparatus would need to be placed in series with the error signal e(n) supplied as an input to filter 12 in order to modify that error signal through objective speech analysis.

As discussed above, the Hollier test apparatus is formed of signal generator 7 and perceptual analyzer 8. Inasmuch as the purpose of the generator is to provide a test signal applied as input to telecommunications equipment under test, i.e. as a reference signal, then signal x(n), being the far-end speech values applied in a forward direction, can be viewed as such an input signal. Hence, signal generator 7, as shown, can be omitted. In this case, signal x(n) will be applied as one input to input ports 6 of analyzer 8. The other input to analyzer 8 would receive the existing signal that has been applied as input to filter 12, namely error signal e(n). There is simply no other signal used in this embodiment to control the operation of filter 12.

What the Examiner had clearly not appreciated before but should now is that a perceptual analyzer will simply not function correctly with such disparate input signals. A perceptual analyzer -- such as that disclosed by the Hollier patent as well as that used by the Applicants, bases its objective measurements on an analysis of a modified

speech signal relative to an incoming or reference speech signal (those modifications here being, e.g., other signals that result from transport of the incoming signal in a forward direction through a telecommunications channel and appear in the incoming signal as it travels in a return path). Both the modified and the incoming speech signals must be applied to the analyzer. That is just NOT the case here. In the proposed arrangement shown in Exhibit B, there is no return signal, in the context of a modified speech signal, applied to the perceptual analyzer that results from passage of far-end speech signal x(n) through the telecommunications channel. Instead, the analyzer will be comparing speech signal x(n) against an error signal e(n) -- two very different and unrelated signals. Error signal e(n) is not a return version of speech signal x(n). Since analyzer 8 has, as input, no return signal based on speech signal x(n) on which to base its measurements, the analyzer will simply not produce an accurate objective estimate of signal quality at all, much less of any return signal. This, in turn, will cause serious adaptation errors in filter 12 -- most likely causing it to diverge, and hence failing to improve both its performance and compensation for incoming signal degradation to any better extent than what would occur without use of the Hollier apparatus (i.e. that shown in Exhibit A).

The very same conclusion occurs for the embodiments shown in Exhibits C and D, where in a very similar fashion, the Hollier test apparatus, specifically analyzer 8, has been incorporated into the embodiments shown in FIGs. 3 and 4, respectively, taught by the Karlsen

application. In each of these instances -- just like that in Exhibit B, analyzer 8 is analyzing signal x(n), NOT against its return signal, but rather against an error signal e. Given this, the same faulty operation of adaptive filter 12 will occur in each instance.

Consequently, the Examiner's proposed approach based on combining the teachings of the Karlsen et al and Hollier patents will fail -- clearly not yielding the results which the Examiner had anticipated.

Hence, if any one of skill in the art were to consider the teachings of the Karlsen et and Hollier applications and attempt to combine those teachings as the Examiner purports to do, that person would quickly realize the adverse results that would likely follow and hence the futility of that approach. Consequently, making the combination as posed by the Examiner would indisputably not lead that person of skill any closer to the present invention than otherwise. Moreover, that person would be further disinclined to combine these teachings to solve the problem the Applicant faces; namely, providing an accurate assessment of objective speech quality at a talker's location, inasmuch as both the Karlsen et al and Hollier patents are directed to providing such assessments at a different location -- that of a listener.

The Applicant's present invention is directed to measuring a so-called *talking quality* of a telephone link in a bi-directional channel (having forward and return directions) in a telecommunications network, e.g., the

influence of echo on the perceptual quality of returned speech *on the talker's side* of that link -- not the listener's side.

As such, it has been left for the Applicant and only the Applicant to discover and teach a proper approach for obtaining an objective measurement of perceptual talker speech quality in a telecommunications channel.

Specifically, the inventive approach, which is diagrammatically shown for comparison purposes in Exhibit E, relies on applying not only an original speech signal as one input to a perceptual speech analyzer but also a combined signal to another input of that analyzer, where the combined signal is formed of a combination of both a returned artifact signal and the original speech signal. Applying both the original talker signal (appearing in the forward channel) and the combined signal, including both that talker signal and the returned artifact signal resulting thereby, to separate inputs of a perceptual speech quality measurement system (PSQM) yields a system that can objectively and accurately determine a quality measurement of the returned signal, as would be perceived by a human listener, relative to the original talker signal.

The use of such a combined signal is simply not taught, disclosed or suggested, however implicitly, in either the Karlsen et al or Hollier applications.

Inasmuch as the teachings of alleged combination fall WELL SHORT of the present inventive teachings, and

specifically the use of such a combined signal, the present invention is clearly not rendered obvious by the teachings of the Karlsen et al and Hollier patent applications regardless of whether those teachings are taken singly or in any combination, including that posed by the Examiner.

Claim 21, as it now stands, contains suitable recitations directed to the distinguishing features of the present invention. Specifically, this claim recites as follows, with those recitations being shown in a bolded typeface:

> "A method for measuring talking quality of a telephone link in a telecommunications network, the method comprising the steps of:
>
> **combining an original talker speech signal and a returned artifact signal, which occurred in a return channel of the telephone link as a consequence of the transmission of the original talker speech signal in a forward channel of the telephone link, to yield a combined speech signal; and**
>
> **subjecting the combined speech signal with respect to the talker speech signal to an objective measurement technique for measuring perceptual quality of speech signals**; and
>
> producing an output signal which represents an estimated value of the talking quality." [emphasis added]

Each of the Applicant's other independent claims 37, 41, 42, 50 and 54 contains similar distinguishing recitations. Moreover, each of these claims precisely defines the returned artifact signal as being the result of transmission of an original talker speech signal applied in a forward channel of the telephone link, thus sufficiently distinguishing the returned artifact signal from an error signal. That

recitation (shown in a bolded typeface below) is clearly evident in claim 37, which recites:

> "A telephone-link circuit for a telephone link in a telecommunications network, wherein the telephone-link circuit has a forward channel and a return channel and an echo-minimizing device included between the forward channel and the return channel, the telephone-link circuit further comprising:
>
> a signal combiner provided with first and second signal inputs which are coupled to the forward channel and the return channel of the telephone link, respectively, and having a signal output which provides a combination of the first and second signals, **wherein the forward channel provides an original talker speech signal and the return channel provides a returned artifact signal which occurred in the return channel of the telephone link as a consequence of transmission of the original talker speech signal in the forward channel;** and
>
> an objective measurement device, provided with a first input port coupled to the forward channel and a second input port, coupled to the signal output of the signal combiner, and an output port, for processing a first original speech signal received on the first input port with the signal output from the signal combiner received on the second input port and for producing an output signal on the output port, the output signal representing an estimated value of taking quality of the telephone link." [emphasis added]

Further, independent claims 42, 50 and 54 each recites a additional distinguishing feature of the present invention, namely, applying the output quality signal to a control input of an echo canceller. Illustratively and in that regard, claim 42 recites as follows, with this further distinguishing feature being indicated by a bolded typeface:

> "A method for measuring talking quality of a telephone link in a telecommunications network, the method comprising the steps of:

combining an original talker speech signal and a returned artifact signal to yield a combined speech signal, the returned artifact signal having occurred in a return channel of the telephone link as a consequence of transmission of the original talker speech signal in a forward channel of the telephone link; and

subjecting the combined speech signal with respect to the original talker speech signal to an objective measurement technique for measuring perceptual quality of speech signals; and

producing an output signal which represents an estimated value of the talking quality and **feeding the output signal as a control input to an echo-minimizing device included in the telephone link.**" [emphasis added]

Consequently, none of the independent claims, now pending in the application, is rendered obvious under the provisions of 35 USC § 103 by the teachings in the Karlsen et al and Hollier patent applications -- regardless of whether those teachings are taken singly or in any combination, including that put forth by the Examiner.

Accordingly, all the pending dependent claims, each of which recites further distinguishing features of the present invention, are not rendered obvious for the exact same reasons set forth above with respect to their corresponding independent claims. Thus, all these dependent claims are patentable as well.

Conclusion

Consequently, the Applicant believes that all these claims are presently in condition for allowance. Both

reconsideration of this application and its swift passage to issue are now earnestly solicited.

Respectfully submitted,

August 11, 2006

Peter L. Michaelson, Attorney
Reg. No. 30,090
Customer No. 007265
(732) 542-7800

MICHAELSON & ASSOCIATES
Counselors at Law
P.O. Box 8489
Red Bank, New Jersey 07701-8489

CERTIFICATE OF MAILING under 37 C.F.R. 1.8(a)

I hereby certify that this correspondence is being deposited on **August 14, 2006** with the United States Postal Service as first class mail, with sufficient postage, in an envelope addressed to the Mail Stop RCE, Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.

30,090

Signature Reg. No.

(PTT133RCEPREAMEND081106/ca:Sitka)

EXHIBIT A

FIG. 2 – Karlsen et al application (WO 97/15124)

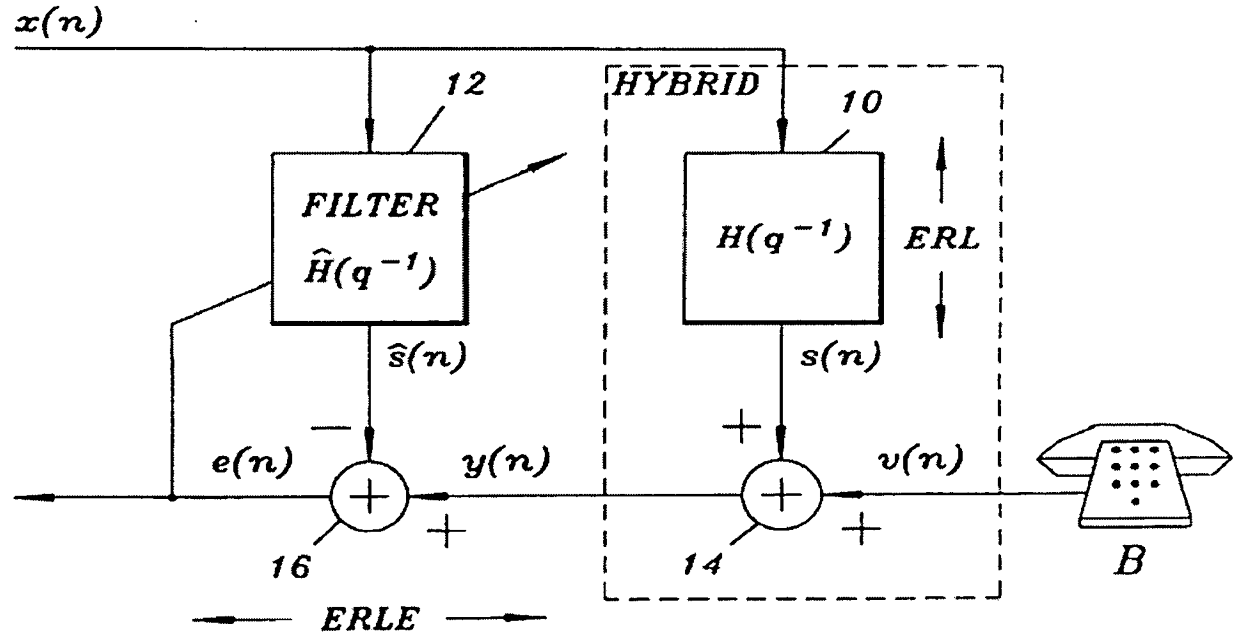


EXHIBIT B

FIG. 2 – Karlsen et al application (WO 97/15124) modified by addition of test apparatus taught by Hollier application (WO 94/00922)

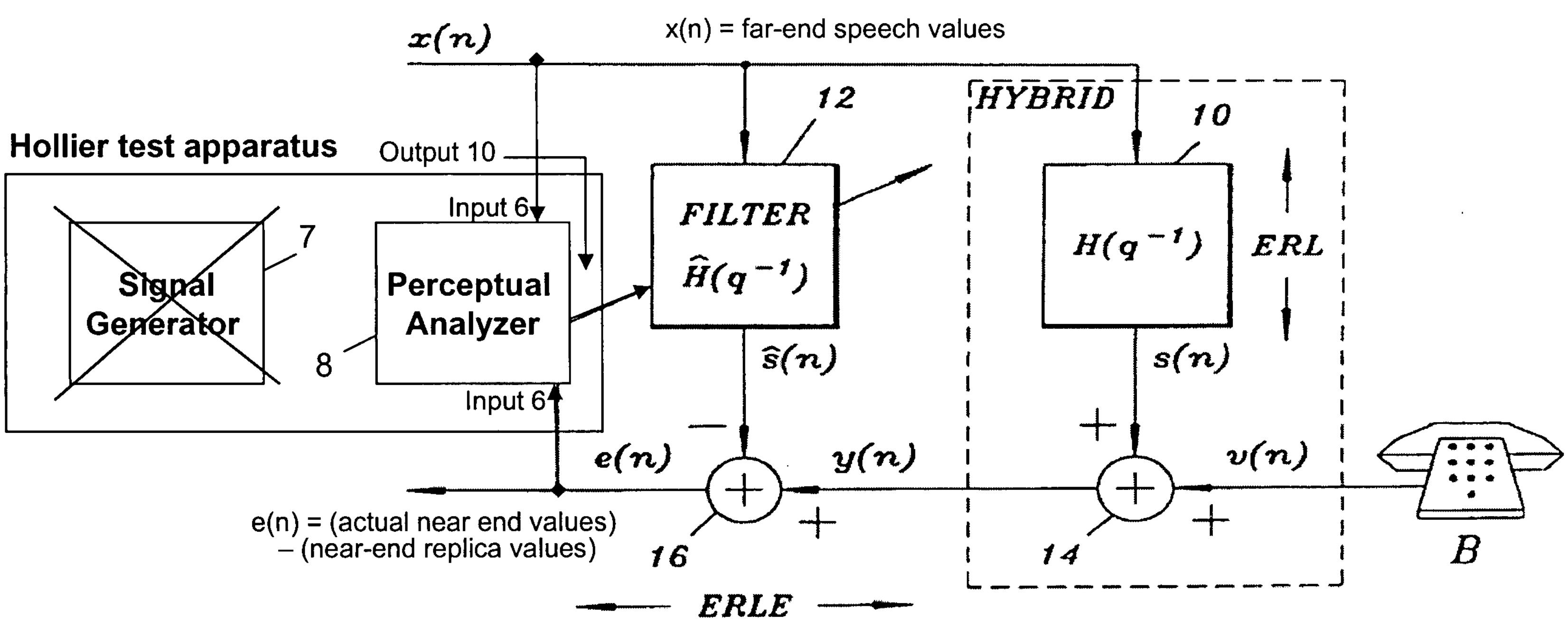


EXHIBIT C

FIG. 3 – Karlsen et al application (WO 97/15124) modified by addition of test apparatus taught by Hollier application (WO 94/00922)

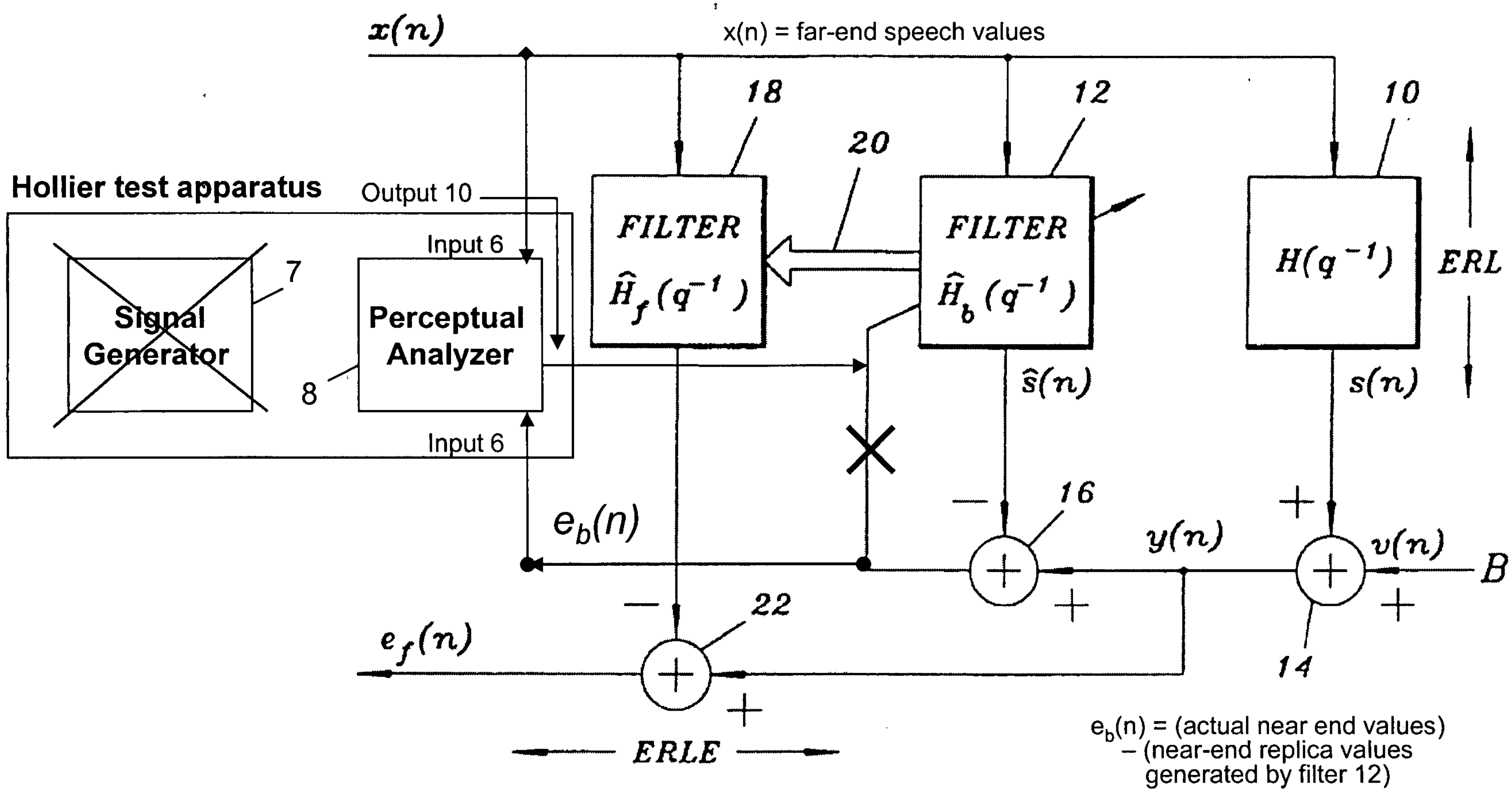
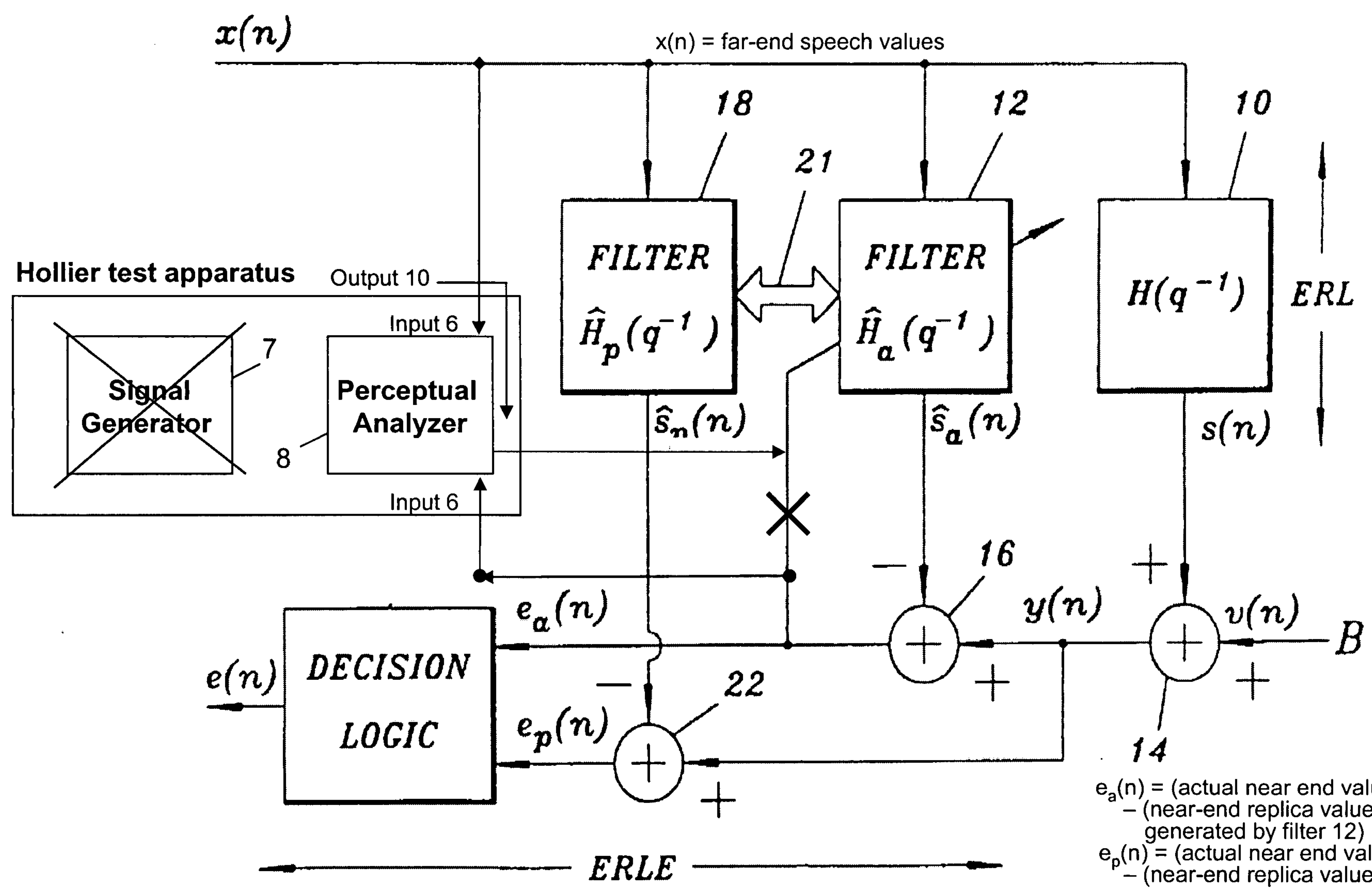


EXHIBIT D

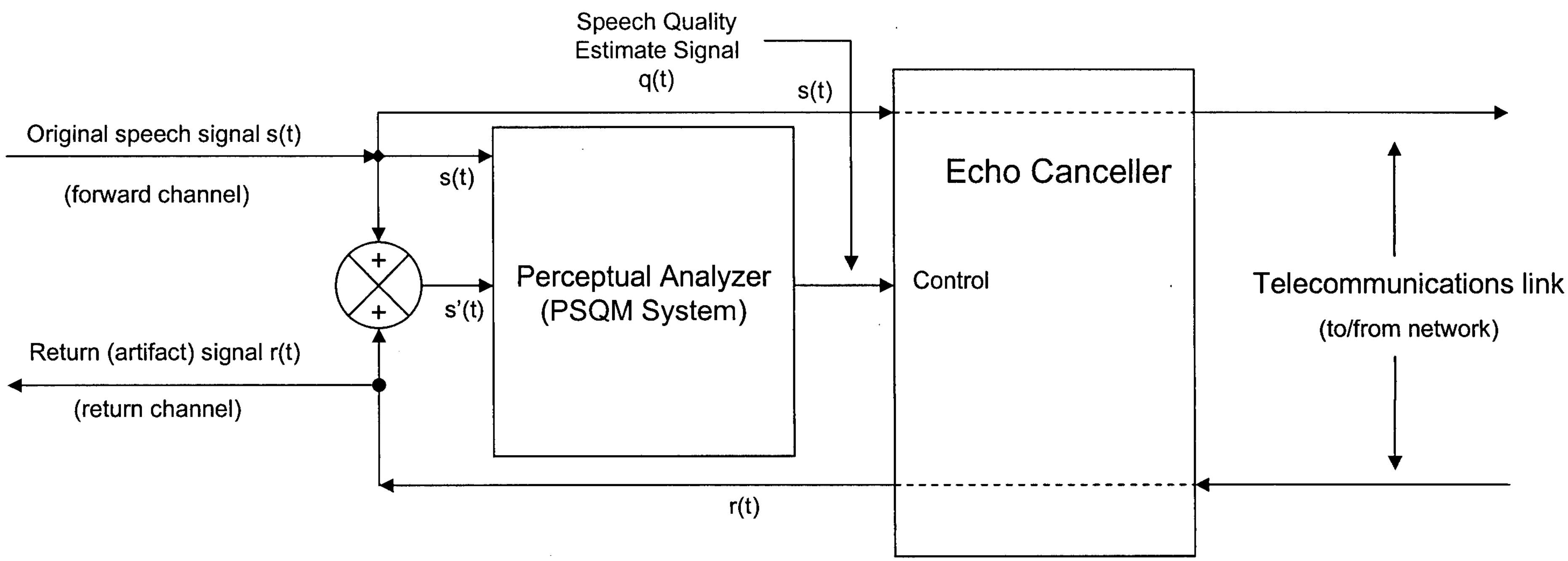
FIG. 4 – Karlsen et al application (WO 97/15124) modified by addition of test apparatus taught by Hollier application (WO 94/00922)



$e_a(n)$ = (actual near end values) – (near-end replica values generated by filter 12)
$e_p(n)$ = (actual near end values) – (near-end replica values generated by filter 18)

EXHIBIT E

Present Invention used with Echo Canceller



s'(t) = original speech signal + return signal
(i.e. s(t) + r(t))